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(54) Title: METHOD AND APPARATUS FOR ELECTRICAL SIGNAL CODING

Encoder — Decoder

(57) Abstract

In a method and apparatus for coding high quality digital audio signals, the signal is sampled at repeated intervals and the sample is filtered into a number of sub-bands, preferably four. The signal in each sub-band is coded by means of a linear prediction technique and each sub-band is quantized with a varying number of levels according to its signal variance. This enables the considerable spectral redundancy which occurs in music signals to be exploited, and this advantage is increased as compared with conventional systems as the signal spectrum becomes discontinuous and/or resonant which is a common occurrence with music.

-1-

METHOD AND APPARATUS FOR ELECTRICAL SIGNAL CODING

The present invention relates to a method of and
apparatus for electrical signal coding.

5

The rapid evolution of the Integrated Service Digital
Network (ISDN) has prompted a growing interest in the
efficient transmission of high quality music signals.
However in contrast to advances made in the field of
10   low-bit rate speech coding, existing real-time music
coding schemes, such as digital companding and delta
modulation, barely attain bit rates below 400 kbit/s, and
are thus incompatible with the 128 kbit/s ISDN capacity
standard. These systems, unlike their speech
15   counterparts, exploit little if any of the natural
redundancies associated with the sound signals of
interest. This situation has remained - due partly to the
higher sampling overheads involved, the absence, until
recently, of high speed digital signal processing hardware
20   and the widely held view that high quality music signals
simply could not be transmitted with low bit rates.

According to one aspect of the present invention there
is provided a method of electrical signal coding
25   comprising the steps of sampling the signal at repeated
intervals, filtering each sample into at least two
frequency sub-bands, coding the signal in each band by means

-2-

of a linear prediction technique and quantizing each
sub-band with a varying number of levels according to its
signal variance.

5   According to another aspect of the present invention,
there is provided apparatus for the coding of an
electrical signal comprising, means for sampling the
signal at repeated intervals, means for splitting the
signal into at least two frequency sub-bands, means for
10   coding the signal in each band by means of a linear
prediction technique and quantizing each sub-band with a
varying number of levels according to its signal variance.

It is well known that the conventional music coding
15   techniques such as companded pulse code modulation (PCM)
and Adaptive Delta Modulation (ADM) are particularly prone
to noise modulation in the presence of monophonic type
signals. This problem is related to the poor masking
properties of these signals and the fact that quantization
20   is singularly performed on the entire spectra. The degree
of noise modulation (or noise breathing) is generally
accepted to be the ultimate limiting factor in the
performance of these coders,leading to usable data rates
of the order of 400 kbit/s. Since sub-band ADPCM music
25   coding bears some similarity to companding and delta
modulation schemes, by the virtue of quantizing in the
time domain, it is reasonable to assume that noise
modulation effects are likely to predominate in sub-band

-3-

ADPCM at low data rates. However, since other fundamental
differences exist between the techniques, in practice the
effects of noise and distortion in sub-band ADPCM are
substantially lower. This fact leads to a considerable

5   coding gain over PCM based systems and the provision
therefore to reduce the bit rate for comparable
performances. The main coding mechanisms involved in
sub-band ADPCM, and the perceptual advantages related to
the underlying characteristics of music signals are as

10   follows:-

1)   With the sub-band coding approach the music is split
      into several frequency bands prior to quantization.
      Modulated quantization noise (which is inherent with

15   adaptive PCM) developed at the coding stages is
      therefore constrained to each band and cannot interfere
      with signals in any other band. The advantage of this
      is that noise masking by the dominant inband signal is
      much more effective due to the reduction in the noise

20   bandwidth. With the music spectrum located in several
      independent bands, the opportunity arises to exploit
      the vast spectral redundancy which occurs in music
      signals by quantizing each band with a varying number
      of levels, according to its signal variance. This

25   technique normally results in a direct objective gain
      over PCM, becoming more advantageous as the signal
      spectrum becomes discontinuous and/or resonant - a very
      common situation in music.

-4-

2) The appropriateness of linear prediction techniques for
the coding of speech signals has in the past been well
documented. The inventors studies have shown that
theoretical short term prediction gains for wideband

5   musical sounds can, in some instances, rise as high as
50 dB. This implies that as many as 8 bits in a
predictive coder can be dispensed with to produce noise
levels comparable to PCM. One particular advantage of
linear prediction is that its efficiency rises with

10   signal periodicity, or spectal purity, and therefore
possesses the ability to directly attenuate, prior to
quantization, those signals which normally promote
audible noise modulation.

15 3) Backward adaptive quantization has some similarities
with instantaneous digital companding, except that the
quantizer input range adjusts according to the size of
past output samples, providing an optimal quantizer
match over a wide music dynamic range. Moreover, since

20   the signal to noise ratio (SRN) is in effect fixed,
background noise under idle conditions is inaudible.
The backward adaptation approach avoids the need to
transmit gain information, unlike block companding, and
is inherently more robust to transmission errors.

25

    In order that the invention may be more clearly
understood, an embodiment thereof will now be described, by

-5-

way of exmaple, with reference to the accompanying drawing,
in which:-

Figure 1 shows a 2-bit pulse code modulation quantizer

5   characteristic,

Figure 2 shows an adaptive pulse code modulation
characteristic,

10   Figure 3 diagrammatically illustrates a backward
adaptive pulse code modulation system,

Figure 4 diagrammatically illustrates a differential
pulse code modulation system,

15

Figure 5 shows a comparison of a pulse code modulation
system and a differential pulse code modulation system,

Figure 6 shows a sound signal spectrum,

20

Figure 7 shows the spectrum of Figure 6 coded with 4
bit pulse code modulation,

Figure 8 shows the spectrum of Figure 6 split into two

25   sub-bands,

Figure 9 shows the spectrum of Figure 8 coded using 4
bit pulse code modulation in each sub-band,

-6-

Figure 10 shows the spectrum of Figure 8 coded using optimal 6:2 bit allocation in the respective sub-bands,

Figure 11 shows adaptive bit allocation in accordance
5   with sub-band energy, and

Figure 12a shows a diagram of a 2-band sub-band ADPCM music coder,

10   Figure 12b shows the 2-band 24-tap QMF tree structue for the coder of Figure 12a,

Figure 13a shows a diagram of a 4-band sub-band ADPCM music coder,

15

Figure 13b shows the 4-band 64-tap QMF tree structure for the coder of Figure 13a.

Figure 14 is a simplified block circuit diagram of
20   one form of coding apparatus using a compact disc player,

Figure 15 represents sub-bands ADPCM encoding in the apparatus of Figure 14.

25   Figure 16 represents sub-band ADPCM decoding in the apparatus of Figure 14.

-7-

Figure 17 shows interfacing for parts of the apparatus
of Figure 14.

Figure 18 shows more detail for the interfacing of
5   Figure 17, and

Figure 19 shows a typical word format for the data
transmission between parts of the apparatus of Figure 14.

10   Pulse Code Modulation (PCM) is a digital coding system
which converts an analogue time waveform into a digital bit
stream consisting of 0's and 1's. To do this, the analogue
signal is first sampled, i.e. the voltage excursion of the
signal is periodically measured, and each analogue sample
15   converted into a binary number, usually in 2's complement.
This binary number represents the size and polarity of the
sample voltage. Each binary number is then transmitted or
stored, instead of the original analogue signal. According
to the Nyquist theorem, the analogue waveform must be
20   sampled at a frequency at least twice that of the maximum
signal deviation to avoid aliasing distortion. Assuming
the sampling criterion is maintained, inaccuracy in a PCM
system can only result through the binary representation of
each sample voltage. For high quality digital audio, it is
25   generally accepted that a binary word length of
approximately 14 to 16 bits is required for this purpose.
The audio signals with which we are concerned would
normally lie in the range 0 - 24 KHz.

-8-

To illustrate the PCM process, figure 1 shows a 2 bit
PCM quantizer characteristic. Using a 2 bit binary word
for the sample coding, i.e. to represent the polarity and
amplitude, results in a quantizer with only 4 levels.
5   Figure 1 also assumes that the maximum input or sample
voltage will not exceed +/-1 volt (i.e. the quantizer
dynamic range is set to +/- 1 volt).

If a sample of +0.6 volts is applied to this 2 bit
10   quantizer, for example, the sample will be represented by
the PCM word 01, using the x-axis. The process of
allocating discrete levels to an analogue voltage is known
as quantization. When this word is transmitted to a
receiving PCM system, the sample is reconstructed to +1.0
15   volt (using the y-axis). This process of reconstruction is
known as inverese quantization.

Since the reconstructed samples can only have 4
discrete levels, PCM coding using only 2 bits introduces
20   gross error, or distortion, with respective to the original
signal. In the example the step-size of the quantizer of
0.5 volts. It follows that the maximum error which can be
incurred by the PCM process is +/-0.25 volts, that is half
the step-size, and therefore the maximum signal to error
25   ratio (SNR) is 1.0/0.25 (or 12dB in the log domain).

A number of characeristics resulting from PCM coding

-9-

may be summarized as follows:-

1) The Root Mean Square (RMS) of the error is independent

    of the input signal amplitude, that is the step-size

5   and therefore the average error is fixed. It is

    paramount therefore that the quantizer dynamic range is

    scaled such that it encompasses the likely signal

    excursion range, to maintain as high a SNR as possible.

10 2) For a reasonably fine quantizer characteristics the PCM

    sample error is perceived as random noise of constant

    average amplitude. The frequency content of this noise

    therefore extends from d.c. to half the sampling

    frequency.

15

    3) Since the maximum error amplitude is half of the

    step-size, doubling the number of levels while

    maintaining the same quantizer range, will halve the

    error with respect to each sample. In a dB scale this

20 reduction in the error voltage represents a 6dB gain in

    the SNR for every additional bit added to the PCM word

    length. For example, a 16 bit quantizer can maintain

    SNR figures in the region of 96dB for full scale input

    samples (that is 16X6dB). A 15 bit PCM system will

25 atain figures in the range of 90dB etc.

    Standard PCM, as described above, exhibits 2 fundamental

    operational disdvantages.

-10-

1)   the input signal must not exceed the quantizer range,
     that is the sample amplitude should not be greater than
     the last quantizer step (figure 1). If this happens,
     the error voltage is no longer constrained to half the
5   step-size. Even with the finest quantization this will
     produce enormous distortion.

2)   The input signal should not become so small that the
     quantizer noise voltage becomes comparable. The SNR of
10   a fixer range quantizer is at its lowest for very small
     input samples.

     In practice, to avoid (1) and (2), the quantizer
step-size is made very fine by using a large binary word
15   length, and the dynamic range of the levels set to twice
that expected of the input signal (that is only
the bottom half of the steps is used). However, if the
quantizer range is allowed to adapt according to the energy
of the input, the use of very long PCM words may be
20   avoided. Music signals have been successfully coded with
adaptive PCM schemes using word lengths around 10 to 12
bits per sample. Adaptive PCM is also known as Digital
companding, where, in this system is the dynamic range of
the input signal that is modified rather than the
25   quantizer. Such a system is illustrated in Figure 2.

     If the quantizer range now matches that of the signal

-11-

at all times, the ratio of the signal voltage with respect
to the noise voltage is constant. In other words the noise
level is directly proportional to the signal level, unlike
PCM. The actual noise level with respect to the signal is

5   simply determined by the number of levels in the quantizer.
If the quantizer range is adjusted at the transmitter to
match the incoming sample amplitudes, likewise the range of
the inverse quantizer must also be adjusted for correct
operation. This may be achieved in two ways:

10

1)   Block adaptation
With this process a block of incoming samples is
delayed at the receiver and the RMS energy of the
block calculated. This energy value is then used to

15   scale the quantizer prior to quantizing the block of
samples. The energy sample is also transmitted to the
receiver ahead of the resulting binary bit stream, in
order to match the inverse quantizer range to that in
the transmitter.

20

2)   Backward Adaptive PCM, which is illustrated in Figure
3 is a system in which the energy is past inverse
quantized samples is used to adjust the range for the
next sample. Since each quantized sample bears

25   information concerning the range adaptation for the
next sample, no extra 'energy' information needs to be
transmitted to the receiver.

-12-

Differential Pulse Code Modulation (DPCM) is illustrated in

Figure 4. DPCM is very similar to PCM except that it is

the amplitude of the difference d(n) between the input

sample x(n) and a predicted sample p(n) which is quantized.

5   P(n) is produced by a predictor P which bases is prediction

on previous reconstructed samples r(n), available at both

the receiver R and transmitter T. Generally the predicted

value is a linear combination of several past reconstructed

samples, according to the equation:

10

$$P(n) = \sum_{i=1}^{\text{Predictor order}} A_i * r(n-i)$$

15

$(A_i = \text{Predictor Coeffs})$

At the receiver, the prediction will always be identical

and therefore the only difference between r(n) and x(n) is

20   the quantization error incurred in quantizing the

difference signal d(n). Because the intention of DPCM is to

reduce the amplitude of the difference signal DS with

respect to the input signal, the range of the quantizer

will on average be smaller than that required to quantize

25   the input, and therefore the error level is reduced on that

found in PCM. DPCM becomes more efficient with respect to

PCM as the predicted signal PS becomes close to the input

samples. Because the prediction is based on previous

-13-

samples, the coding gain exhibited by DPCM is a result of
its ability to exploit signal redundancy in the time
domain. The differences between PCM and DPCM are
illustrated in Figure 5. DPCMs ability to reduce the noise
5   level N may be measured in the log domain for perceptual
reasons. For example, if the noise level NP in DPCM is
halved, compared to PCM, for some signal S, we say the
coding gain over PCM is 6dB. Because the dynamic range of
the difference signal is very much smaller than that of the
10 input signal, DPCM results in less quantization noise.

DPCM is useful only if it is capable of predicting the
incoming signal. For fixed signal characteristics such as
sinewaves, this is relatively easy, and SNR coding gains up
15 to 60dB may be realized. However if the predictor has been
optimized for any particular waveform, its performance will
be sub-optimal for other types of signal. To sum up, if the
input signal to DPCM systems is non-stationary, the linear
combination within the predictor must adapt according to
20 the signal statistics to maximize the coding code. In
practice the predictor coefficients are calculated such
that the means squared error between the predicted signal
and the input is minimized. This calculation can be
performed every sample using gradient or lattice adaptation
25 techniques or on a block of input samples as with the
autocorrelation and covariance methods.

By combining both LPC adaptation for the predictor and

-14-

APCM for the quantizer within the basic DPCM system, what
is known as an Adaptive Pulse Code Modulation (ADPCM)
scheme is arrived at.

5   Looking at the short-term frequency spectrum of a
typical audio sound we find that the spectral components
are not evenly distributed across the spectrum. In
practice most of the signal energy will be found in the
lower half of the spectrum, most of the time, figure 6.
10 However, it is known that the spectrum of the error voltage
generated in the coding process (be it PCM, DPCM or ADPCM)
exhibits a white noise spectrum, extending from dc to half
the sampling frequency. For example, if the signal shown
in Figure 6 is coding using a 4 bit PCM system, the noise
15 energy level will lie approximately 24 dBs below the
average signal level, see Figure 7. However on coding the
spectrum the resulting noise level exceeds that of the
signal for frequencies beyond 7kHz. Subjectively this
coding error will be perceived as high frequency background
20 hiss, since the main signal cannot 'mask' the error signal
at high frequencies. The problem therefore with full-band
PCM is that the noise level may be pushed up by high
energy spectral resonances, to such an extent that it may
'drown out' distant spectral components of much lower
25 amplitude, Figure 7.

        One way of decoupling error signals generated by
different parts of the spectrum is to divide the spectrum

-15-

up to several bands using sub-band coding filter banks,
prior to quantization. If each band is then individually
coded using say PCM,DPCM or ADPCM, the noise generated by
each band cannot interfere with signals in other bands.

5   Using the spectrum of Figure 6, if this is split into 2
equal bandwidth bands we obtain Figure 8. Each band, band
1 and band 2, now contains a signal of only 8 kHz bandwidth
and therefore the sampling frequency within each band can be
halved to 16kHz, rather than 32 kHz for the full-band

10   signal.

Coding each band individually with 4 bit quantizers
results in Figure 9. In this case the noise in the lower
band exceeds the signal only between 7 and 8kHz. However,

15   since the noise level in the upper band is now
unnecessarily low, a better allocation of quantizer might
be that shown in Figure 10. The aim therefore of sub-band
coding is to allow the bit distribution to match that of
the signal variance within each band, and in doing so,

20   maximizing the likelihood of noise masking. Obviously the
spectrum pattern of figure 6 will change with time,
implying that an even greater gain could be achieved by
allowing the bit allocation to adapt to the short-term
energy in each sub-band, figure 11. This process is known

25   as Adaptive Bit Allocation in the frequency domain.

Because the bit allocation can approximate the energy
contour of the signal spectrum, sub-band coding can

-16-

therefore directly exploit the spectral redundancy

associated with almost all signals, to achieve an improved

coding gain over full-band coding techniques.

5   Referring to Figures 12a and 13a the main parts are

referenced as follows:-

        Q   Backward Adaptive Quantizer

        1/Q Inverse Backward Adaptive Quantizer

10   Quantizer Step-Size Adaptor

        P   All-Pole Backward Adaptive Predictor

        ABA Adaptive Bit Allocation

        This music coder consists of a two band Quadrature

15 Mirror Filter (QMF) filter bank incorporating backward

adaptive prediction and quantization in each band. Initial

work with this system was undertaken with a fixed bit

allocation of 5:3 and either block adaptive or gradient

adaptive LPC within in both upper and lower band

20 predictors. The quantizers used are non-uniform Laplacian

pdf backward adaptive. The two bands are divided and

recombined using a 24 tap Quadrature Mirror Filter (QMF).

15kHz audio signals are sampled at 32 kHz and each 16 bit

sample filtered into two frequency sub-bands, coded to 4

25 bits and transmitted to the receiver. Here the compressed

samples are decoded, reconstructed and converted back to

analogue. A similar scheme to that shown in Figure 12a

and 12b is shown in Figures 13a and 13b. In these Figures

-17-

the main parts are referenced as follows:

    Q   Backward Adaptive Quantizer

    1/Q   Inverse Backward Adaptive Quantizer

5   P$_{(PZ)}$ Pole-Zero Gradient Adaptive Predictor

           Quantizer Step-Size Adaptor

    QMF   Quadrature Mirror Filter

        In this arrangement, there are 4-sub-bands. The

10 increase in the number of sub-bands increased the quality of

    the coded music still further as compared with the scheme

    of Figures 12a and 12b but at the cost of increased

    complexity. The coder uses a 4 band 64 tap QMF tree

    structure (0-4 4-8 8-12 12-16kHz) where each band

15 incorporates an 8 pole backward block adaptive predictor

    (updated using the autocorrelation method) and a backward

    adaptive Laplacian quantizer. In this coder however, the

    bit allocation is allowed to adapt in a backward mode,

    according to the short term energy distribution of the

20 error signal across the 4 bands. The liberal use of

    backward adaption in this coder also avoides the need to

    transmit SI. However the coding delay is now 192 samples,

    or 6.1mS and the overall complexity approximately four

    times that of the 2 band scheme.

25

        The tree structures of Figures 12b and 13b show the

    splitting of the Input Audio signal for the front end of

    the encoder. To reconstruct the sub-bands at the decoder,

-18-

the signal directions in the tree structure diagrams would
be reversed and the final signal would be the AUDIO OUT.

Vitally important to the successful evaluation of any
5   high quality coding scheme is the maintenance of signal
fidelity throughout all the audio acquisition, coding and
playback processes. This is important if delicate coding
imperfections are to be reliably revealed and their effects
assessed. All of the source material used in the
10   subjective assessments was obtained from either a CD player
of a PCM-F1 digital recorder. Digitization of the music
signal was achieved using a purpose built data acquisition
unit, employing 16 bit stereo A to D and D to A converters.
Anti-aliasing and reconstruction filtering was performed
15   with 9th order 16 kHz roll-off active filters. To evaluate
the success of this coding arrangement, coded music was
compared to the original (bandlimit to 15kHz), by listening
to both passages through high quality loudspeakers, placed
in a room considered to be representative of a good
20   domestic environment. The playback equipment consisted of
the PCM-F1 recorder, an active crossover network with
attenuator, four 100 watt reference mono power amplifiers
and a pair of bi-amped monitor quality loudspeakers. A
minimum of audio equipment was used at all times to reduce
25   cumulative signal degradation.

The results obtained clearly demonstrated the very
high coding efficiency of sub-band ADPCM with comparison to

-19-

digitally companded and ADM schemes when applied to music

signals. The results show that surprisingly high quality

digital audio can be represented by split band ADPCM words

approaching one quarter that of linear PCM, and still

5   remain essentially indistinguishable. The inventor

believes these findings are very significant and represent a

major advance in digital audio technology, having immediate

implications for ISDN, broadcasting and DBS digital audio

distribution.

10

      Whilst the above has dealt with 4 bit sub-band ADPCM,

for the purposes of low-bit rate music coding, auxiliary

tests have shown that the performance curve for this scheme

saturates at approximately 5-6 bits per Nyquist sample,

15   rendering even the most critical music signals

indistinguishable from the original.   In addition to this

ADPCM is known to offer bit error immunity down to 1 to

10^3 without any form of protection or concealment - 4 or 5

orders of magnitude better than any PCM based systems.

20   With future professional digital audio standards looking

beyond 16 bits per sample as a means of improving dynamic

capability. The continuing viability of PCM as a reliable

and economic means of storing or transmitting high quality

digital audio must be open to question.

25

      Referring to Figure 14 a system is described which

demonstrates an application of the signal coding technique

using digital audio samples derived from a Compact Disc

-20-

(CD) player 141. With this system, the 16 bit PCM audio

data stream within the CD player is firstly compressed to 4

bits and then expanded back up to the original 16 bit PCM

format, returned to the player circuitry and converted to

5   analogue as normal. The compression/expansion (or coding)

processes are performed in realtime using to separate

digital signal processor (DSP) chips 142 and 143. Since

both the encoding and decoding process is performed

back-to-back, use of the very high quality audio material

10   provided by the compact disc playback system allows for an

accurate evaluation of the sonic properties of the coding

system as a whole.

The coding apparatus was programmed into two AT & T

15   DSP16-55nS digital signal processor chips 142, 143, which

were in turn interfaced to a modified Philips CD650 compact

disc player 141 (the disc is referenced 140).

The CD 650 player digital audio circuits consist of

20   three main PCM processing chips the demodulator 144

(SAA7210 or the A-Chip), the Digital Filter 145 (SAS7220 or

the B-Chip) and the Digital to Analogue Converter 146

(DAC). In order to acquire the 16 bit PCM audio data, the

normal serial data path between the SAA7210 and the SAA7220

25   is broken and alternatively re-routed via the

encoder-decoder hardware processing loop 147. On inputting

each 16 bit PCM sample (sterio sampling rate = 88.2 kHz)

the encoder effectively transfers coded data serially to

-21-

the decoder at a quarter of this bit rate (Eg 4 bits per
PCM sample). On receiving the compressed data, the decoder
chip 143 expands it back to 16 bits PCM and outputs the PCM
data stream to the SAA7220 filter chip 145 in a similar
5   format to that of the demodulator 144 (SAA7210 PCM) output
data stream.

Referring to Figure 15, on entering the encoding DSP
chip 142 through its serial input port, the 16 bit PCM
10   words are first applied to the sub-band filter bank
algorithm 150, which divides the digital spectrum into four
uniform channels or sub-bands, 0-5.5kHz, 5.5-11kHz, 11-16.5Khz
and 16.5-22kHz. Each sub-band signal is then down-sampled
to 11kHz giving a throughput of one word per sub-band for
15   every four 16 bit PCM words input to the filter.
Each sub-band signal is then processed through independent
ADPCM algorithms which subtract a prediction (P) from each
sub-band sample and quantises (Q) the difference. The
quantiser bit allocation in this illustration is fixed at 8
20   bits for the 0-5.5kHz band, 4 bits for the 5.5-11kHz band
and 2 bits each for the remaining bands. The four quantised
difference signals are then multiplexed by multiplexer 15
into a single 16 bit sub-band ADPCM word (ie 8+4+2+2 bits)
Since this composite word is formed for every four
25   PCM words input to the encoder chip 142, by storing or
transmitting each sub-band ADPCM word instead of the
original PCM words, a four fold reduction in signal
capacity is facilitated.

-22-

Each 16 bit sub-band ADPCM word is therefore output
serially, through the serial output port of the processor,
to the decoder DSP, chip 143. As explained in the previous
paragraph, these exchanges are now at a quarter of the rate
5   at which the 16 bit PCM words are entering the encoder chip
142.

Referring to Figure 16, on entering the decoding DSP
chip 143 through its serial input port, the 16 bit ADPCM
10   words are firstly applied to the demultiplexer algorithm
160. Once the sub-band codes have been separated, each are
inverse quantised (1/Q) and reconstructed by adding back
each prediction (P) subtracted at the encoding chip 142.
The reconstructed sub-band signals are then up-sampled and
15   inverse filtered back into a 44.1kHz sampled sterio PCM
signal. Each PCM word is then ouptut to the Digital Filter
145 (SAA7220) via the DSP's serial output port.

While the left and right channel PCM samples are input
20   to the encoder through a common port, each are encoded and
decoded separately throughout the back-to-back coding
process. The decoder and encoder algorithms 160 and 150
therefore incorporate in practice, two of the routines
shown in Fig. 16 and Fig 15 respectively to process the
25   left and right channels.

Compact Disc Player - DSP16 Hardware Interfacing

-23-

Progressively detailed description of the hardware
interfacing for the chips 142 and 143 is given with
reference to Figs.17 and 18 respectively.

5   Starting at the PCM serial input of the encoder
processor (Data Input DI), PCM audio data, in the I²S
format (ref.1), is simultaneously clocked out of the
demodulator chip 144 (SAA7210) and into the DI of the DSP
encoder 142. Each encoder 16 bit, serial input is
10   initiated on the edges of WSAB via the load cycle LDAB. In
order for the processor to distinguish between left and
right PCM words, a sync pulse is applied to an "active"
parallel input line (PBO) in order to correspond to the
acquisition of a left PCM sample. On reading both the
15   parallel and serial ports simultaneously, the processor can
correctly orientate the processing of each sample reliably.

The 16 bit sub-band ADPCM words are output from DO
(Data out) of the DSP 16 serial output port. However,
20   these are transferred at a quarter of the rate of the PCM
words. However, the timing left sub-band ADPCM is such that
its entry onto XDAB line corresponds to a SYAB high pulse.
The right channel sub-band ADPCM word follows 45.35µs later,
but without an active pulse on SYAB.

25

Receiving the stereo 16 bit sub-band ADPCM word on XDAB
via DI serial input, the decoder processor simultaneously

-24-

outputs the reconstructed 16 bit PCM words at a rate of

88.2kHz, to the DDAB input of the SAA7220 chip via the

DSP16 DO serial output. Again the output word timing of the

decoding algorithm 160 is such that the left channel PCM

5   words are aligned with a high on the sync input SYAB.

Detailed TTL timing diagrams for the serial and

parallel ports of the AT & T DSP16 processor chips 142 and

143, and the PCM data formats for the Mullard SASA7210 and

10   SAA7220 chips 144 and 145 may be found in appropriate

Mullard and AT & T reference manuals. A typical word

format for the serial 6 bit sub-band ADPCM data

transmissions between the DSP chips 142 and 143 is shown in

Fig 19 i.e. the lower band (0-5.5kHz) ADPCM occupies the

15   first eight bits of the transmitted word, the next band

(5.5-11kHz) ADPCM word occupies the next 4 bits and so

forth.

While the work described here is primarily concerned

20   with high quality music coding for ISDN, applications are

to be found in other bandwidth conscious environments, eg

music distribution for land and satellite-based

broadcasting. Moreover the implications of high quality

music coding are also very significant in the continuing

25   development of the domestic digital audio media, in

particular, sound on video, compact disk (CD), video CD and

digital audio tape (DAT).

-25-

It will be appreciated that the embodiments have
been described by way of example only and that many
variations are possible without departing from the
invention.

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-26-

CLAIMS

1.   A method of electrical signal coding comprising the
steps of sampling the signal at repeated intervals,
filtering each sample into at least two frequency
subbands, coding the signal in each band by means of a
5   linear prediction technique and quantizing each sub-band
with a varying number of levels according to its signal
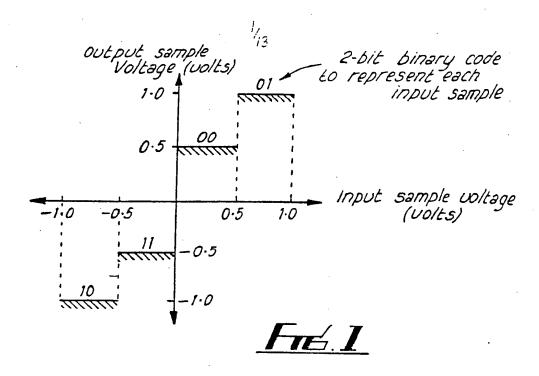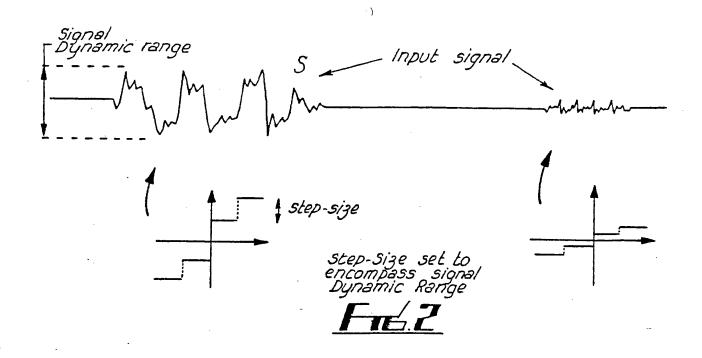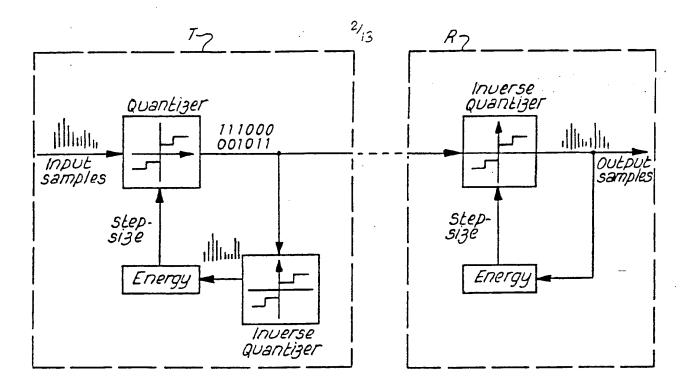variance.

2.   A method of electrical signal coding as claimed in
10   claim 1, in which the signal frequency is in the range
0-24KHz.

3.   A method of electrical signal coding as claimed in
claim 1 or 2, in which each sample is filtered into four
15   sub-bands.

4.   A method of electrical signal coding as claimed in any
preceding claim, in which a block of incoming samples is
delayed at the transmitter, the root means square energy of
20   the block is calculated and the energy value is used to
scale the quantizer prior to quantizing the block of
samples, and the energy sample is also transmitted to the
receiver ahead of the resulting binary bit stream in order
to match the inverse quantizer range to that in the
25   transmitter.

-27-

5. A method of electrical signal coding as claimed in
claims 1 to 3, in which the energy in past inverse quantized
samples is used to adjust the range for the next sample
both at the receiver and the transmitter.

5

6. Apparatus for the coding of an electrical signal
comprising, means for sampling the signal at repeated
intervals, means for splitting the signal into at least
two frequency sub-bands, means for coding the signal in

10   each band by means of a linear prediction technique and
quantizing each sub-band with a varying number of levels
according to its signal variance.

7. Apparatus as claimed in claim 6, in which the means for

15   coding comprises a two band Quadrature Mirror Filter Bank
incorporating backward adaptive prediction.

8. Apparatus as claimed in claim 6, in which the means for
coding comprises a four band Quadrature Mirror Filter Bank.

20

9. Apparatus as claimed in claim 7 or 8, in which block
adaptive predictors are provided in the two sub-bands.

10. Apparatus as claimed in claim 7 or 8, in which

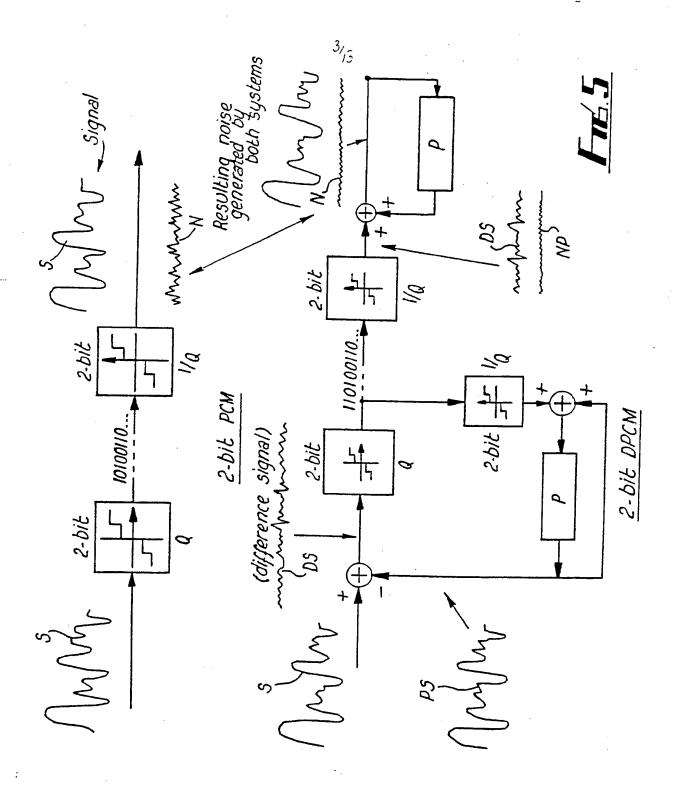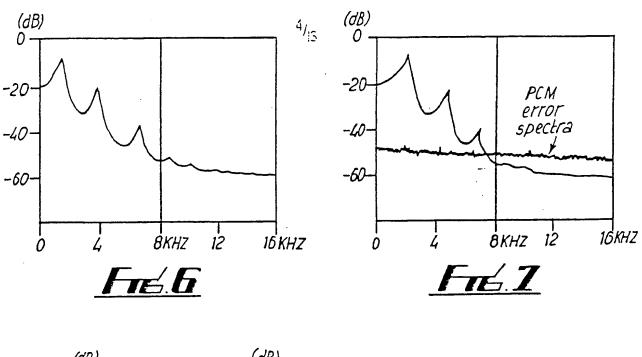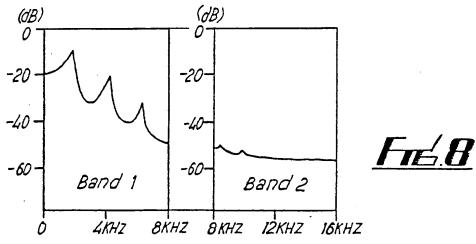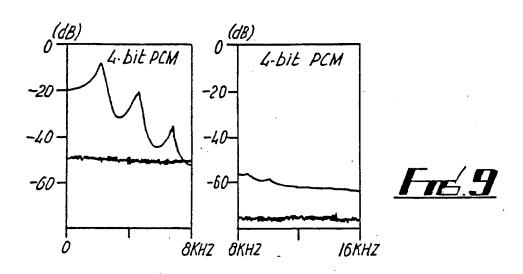25   gradient adaptive predictors are provided in the two
sub-bands.

$^{1}/_{13}$

output sample
Voltage (volts)

2-bit binary code
to represent each
input sample

01

1·0

00

0·5

Input sample voltage
(volts)

−1·0   −0·5   0·5   1·0

11   −0·5

10

−1·0

Fig.1

Signal
Dynamic range

S   Input signal

step-size

Step-Size set to
encompass signal
Dynamic Range

Fig.2

FIG.3

FIG.4

$$P(n) = \sum_{i=1}^{\substack{Predictor \\ order}} a_i * r(n-i) \, — \, — \, — \, — \, — \, \underline{EQN.1}$$

$$(a_i = Predictor \; Coeffs)$$

Fig.5

Fig.6

Fig.7

Fig.8

Fig.9

5/13

Fig.13b

Fig.10

6/13

Fig. 11

7/13

Fig.12a

Fig.12b

Fig.13a

Fig.13b

FIG. 14

16 Bit
PCM words

FS=44·1KHZ

150

Subband Filter Bank

Fig.15

16·5 – 22 KHZ
FS = 11KHZ

11 – 16·5KHZ
FS = 11KHZ

5·5 – 11KHZ
FS = 11KHZ

0 – 5·5KHZ
FS = 11KHZ

2 Bits

2 Bits

4 Bits

8 Bits

M
U
X

16 band
subband
ADPCM words

FS = 11KHZ

151

Fig.16

16 Bit
Subband
ADPCM
words
FS – 11KHZ

160

D
E
M
U
X

2 Bits

2 Bits

4 Bits

8 Bits

16·5 – 22KHZ
FS = 11KHZ

11 – 16·5KHZ
FS = 11KHZ

5·5 – 11KHZ
FS = 11KHZ

0 – 5·5 KHZ
FS = 11 KHZ

Subband Filter Bank

16 Bit
PCM words

FS = 44·1KHZ

11/15

FIG.17

Fig. 18

++ Parallel input in Active Mode
* Processor Interrupt driven by Output Buffer Empty Signal

Fig.19

INTERNATIONAL SEARCH REPORT

I. CLASSIFICATION OF SUBJECT MATTER (if several classification symbols apply, indicate all) *

According to International Patent Classification (IPC) or to both National Classification and IPC

IPC^4: H 04 B 1/66

II. FIELDS SEARCHED

Minimum Documentation Searched [7]

Classification System :	Classification Symbols
IPC^4	H 04 B

Documentation Searched other than Minimum Documentation
to the Extent that such Documents are Included in the Fields Searched [8]

III. DOCUMENTS CONSIDERED TO BE RELEVANT [9]

Category *	Citation of Document, [11] with Indication, where appropriate, of the relevant passages [12]	Relevant to Claim No. [13]
X	IEEE International Conference on Acoustics, Speech and Signal Processing, vol. 3 3,4,5 May 1982 (New York, US) V. Gupta et al.: "Performance evaluation of adaptive quantizers for a 16-Kbit/s sub-band coder", pages 1688-1691, see page 1688, left-hand column, lines 31-41; right-hand column, lines 13-35; page 1689, left-hand column,	1,3,5,6
A	paragraphs 1,2	4,8
	--	
X	IEEE International Conference on Acoustics, Speech and Signal Processing, vol. 4 26-29 March 1985 (New York, US) F.K. Soong et al.: "Subband coding of speech using backward adaptive prediction and bit allocation", pages 1672-1675, see page 1672, left-hand column, first paragraph; page 1675, left-hand column, line 15 - right-hand column,	1,3,5,6
A	line 4	8,10
	--	

* Special categories of cited documents: [10]

"A" document defining the general state of the art which is not considered to be of particular relevance

"E" earlier document but published on or after the International filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the International filing date but later than the priority date claimed

"T" later document published after the International filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.

"&" document member of the same patent family

IV. CERTIFICATION

Date of the Actual Completion of the International Search	Date of Mailing of this International Search Report
9th May 1989	0 8 JUN 1989

International Searching Authority	Signature of Authorized Officer
EUROPEAN PATENT OFFICE	P.C.G. VAN DER PUTTEN

Form PCT/ISA/210 (second sheet) (January 1985)

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III. DOCUMENTS CONSIDERED TO BE RELEVANT   (CONTINUED FROM THE SECOND SHEET)

Category*	Citation of Document, with indication, where appropriate, of the relevant passages	Relevant to Claim No
X	IEEE International Conference on Communications '84, Links for the Future, Science, Systems & Services for Communications", vol. 3, 14-17 May 1984, Amsterdam (NL) (New York, US) R.L. Zinser et al.: "A robust 9.6 Kb/s subband coder design for the Rayleigh fading channel" pages 1163-1168, see page 1163, left-hand column, second paragraph; page 1164, left-hand column, lines 37-67; page 1165, right-hand column, last paragraph	1,3-6
A	Funkschau, vol. 59, no. 13, 19 June 1987, (Munich, DE) G. Theile et al.: "Verdeckungseffekte" pages 44-48, see page 46, middle column, line 6 - page 47, left-hand column, line 8	1,2